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Performance Evaluation to Code-Aided Linear Iterative Equalizer Under a Fading Channel

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Abstract

A linear iterative turbo equalizer (LITE) with code aided scheme is presented to be employed in a fading channels. A method is proposed for exploiting transmittal diversity within a parallel independent fading channel together with an iterative equalizing receiver. The structure of the equalizer combines LITE, encoding, decoding interleaving, and de-interleaving schemes. The code-aided LITE employs encoding and interleaving at the transmitter and passes a priori information on the transmitted symbols between decode outputs at the receiver. The performance of a code-aided LITE is evaluated by means of computer simulation. It shows the proposed code-aided equalizer functions benefits a better system performance in the fading channel.

Keywords

Linear Iterative Turbo Equalizer; Coding Scheme; Fading Channel

Introduction

Imperfect channel degrades the system performance of the wireless communication. In the wireless environment, there are a lot of work developed for effective and inexpensive ways to eliminate these imperfect effects [1]-[3]. An equalizer could be used to mitigate the distorted signal. In order to reduce the degrading effects caused by the fading channel, an equalizer needs to track the channel characteristics, such as the channel impulse response (CIR), in the time-varying wireless channel. Traditionally, the CIR could be estimated by sending a training sequence [4]. Then, the actual data are transmitted later. The method could be called as a sequence aided equalizer. However, the equalizer wastes the transmission bandwidth and it is an ineffective scheme. It needs the training sequence and increases the transmission bandwidth.

In order to increase the bandwidth efficiency, a blind equalizer without a training sequence is proposed to track the time-varying characteristics of the channel. The code-aided equalizer is the one scheme [5]-[7]. The code-aided linear iterative turbo equalizer (LITE) is developed according to the structure of turbo equalization, code-aided adaptive equalizer and error correction coding scheme [8]. The coding scheme improves the performance of the communication system and the decoding sequence is employed as the training sequence to update the adaptive equalizer. It could help the received signal to be recovered.

In this paper, a structure combining with convolutional coding and LITE schemes is proposed. It provides a significant bit error rate (BER) improvement in the communication system. In the following sections, the system model is described. Section III provides the proposed code-aided LITE with convolutional coding scheme and the algorithm used to update the weight of the adaptive equalizer. The results using the proposed LITE are presented in section IV. Finally, conclusions are given in section V.

System Model

The channel characteristics are not fixed but change continuously in the radio propagation environment. Fading

factor depicts the effect on the transmitted signal caused by terrain blockages such as hills, buildings, etc. Generally, a communication system includes the transmitter, the channel and the receiver is shown in Fig. 1.

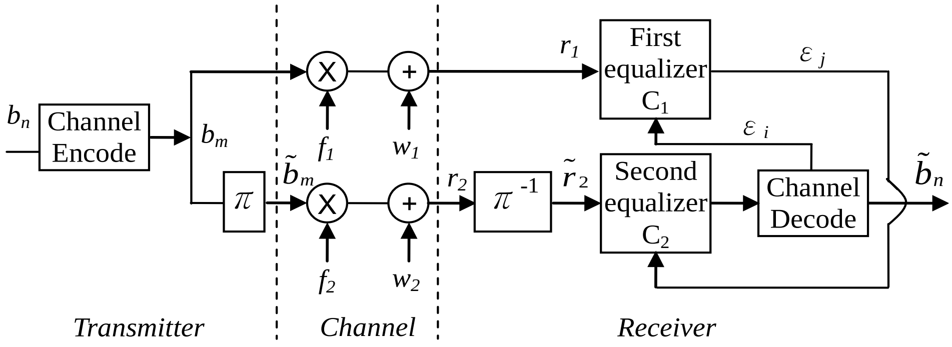

FIG. 1 SYSTEM MODEL IN A FADING CHANNEL

The system considers fading given in the transmission path. In the transmitter, it uses a coding scheme and interleaver. With consideration of multipath transmission, the signal flows through a fading channel. The two channel outputs could be written as

$$r_1(m) = f_1(m) b_m + w_1(m) \tag{1}$$

$$\tilde{r}_2(m) = f_2(m) \tilde{b}_m + w_2(m) \tag{2}$$

where b is the coded signal, is the interleaved coded signal (through interleaver π), f_1 and f_2 are fading factors, w_1 and w_2 are uncorrelated additive white Gaussian noise (AWGN) [8].

At the receiver, the code-aided linear iterative equalizer is employed to equalize the received signal. The equalizer employs a de-interleaver, two equalizers and a channel decoder. At the receiver, the de-interleaver processes the interleaved signal from the transmitter.

The code-aided LITE consists of two stages: the first stage equalizer without decoding procedure but the second one with the benefits of decoding scheme. Initially, the prior information can be represented with a scalar ε_i [8]. The first stage equalizer uses the received vector r1 and the prior information to update its tap coefficients. Then, the second stage equalizer updates its tap coefficients according to the output of the first stage equalizer εj and the received vector . Following the second stage equalizer, the signal is decoded and is used to update the prior information in the first stage equalizer. Iteratively, the updated prior information is processing between the equalizers until reaching a convergence criterion.

Code-Aided Linear Iterative Equalizer

On the one hand,Equalizer helps the distorted signal to be recovered. It improves the performance of a communication system. On the other hand, coding schemes benefit a better performance of the communication system. Combing with equalizer and coding scheme, a code-aided LITE uses a coding scheme is proposed.

The scenario to update the tape weights of the equalizer is given as following. At the receiver, the algorithm for LITE is as

(1) INPUT data: a permutation π, two vectors of received values r_1 and $\tilde{r}_2 = \pi^{-1}(r_2)$, and prior information about the symbols expressed as vector of probabilities ε_{in}^{E1}, where ε_{in}^{E1} is usually a vector containing a value 0.5 in each position for a BPSK data.

(2) Repeat the following steps until a termination criterion (MMSE) is reached.

(a) $\varepsilon_{out}^{E1} = equalization(r_1, \varepsilon_{in}^{E1})$

(b) $\varepsilon_{out}^{E2} = equalization(\tilde{r}_2, \varepsilon_{out}^{E1})$

(c) $\varepsilon_{in}^{E1} = decode(\varepsilon_{out}^{E2})$

(d) If the termination criterion is not satisfied back to step 2(a).

(3) OUTPUT data: the hard decisions for symbols $\tilde{b}$ based on ε_{out}^{E1} and ε_{out}^{E2}.

where E1 and E2 represent the first stage equalizer and second stage equalizer, respectively. εin and εout represent the input of equalizer and output of equalizer, respectively.

Simulation and Results

For the simulation for the code-aided LITE, the coding scheme with convolutional code is employed as an example. The coding rate 1/2 and a structure of (5,7) with minimum Hamming distance of 3 is apoted for the analysis.. The structure includes two registers and modulo 2 adders. Each output sequence does not only depend on the current input information but also on a number of the past information.

In the equalizer, the hardware structure of the finite impulse response filter (FIR) is shown in Fig 2.

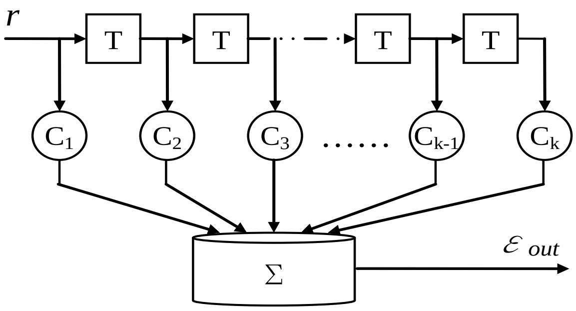

FIG. 2 A FINITE IMPULSE RESPONSE FILTER (FIR)

Consider each of these two equalizers has a 6-tap length. To adjust the tap weight of the equalizer, the equalization algorithm adopts Least Mean Square (LMS). This LMS algorithm is given as

$$C(n+1)=C(n)+\mu e(m)r(m) \tag{3}$$

where $C(n)$ is the tap weight vector of the equalizer at time n, r is the input vector to the equalizer at time n and μ is a constant (0<μ<1).

$$e(m)=\varepsilon-\varepsilon_{out} \tag{4}$$

where ε is the prior information and ε_{out} is the output of the equalizer.

To determine the iterative time for the equalizer reaching a convergence state, the simulation results is shown in fig. 3. In Fig. 3, it shows the average error could be reduced to a stable state after 25 iterative times. Finally, the simulation chooses the iterative times 25 in the works.

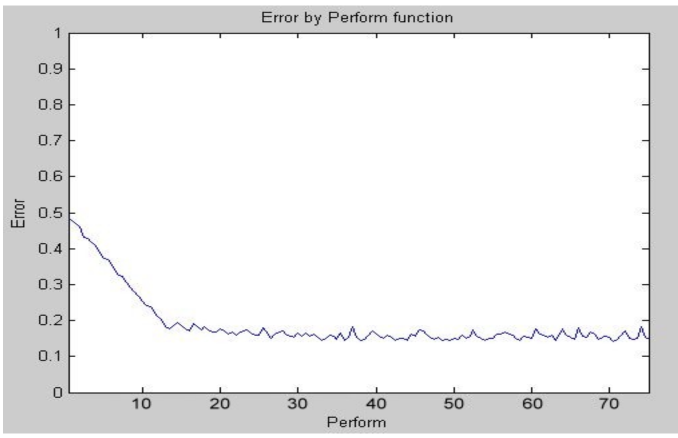

FIG. 3 THE NUMBER OF ITERATIVE TIME

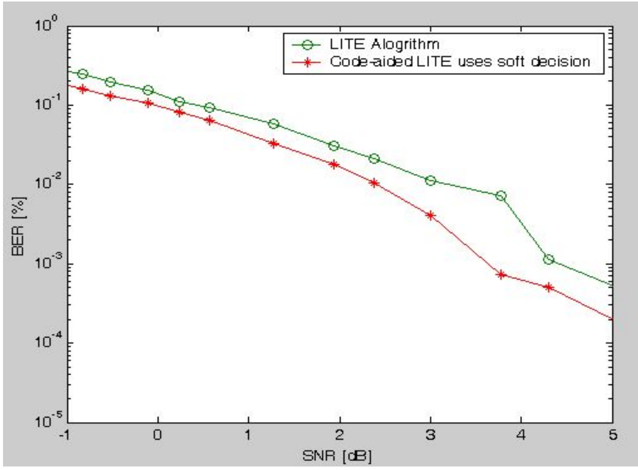

FIG. 4. THE BER PERFORMANCE UNDER A AWGN CHANNEL

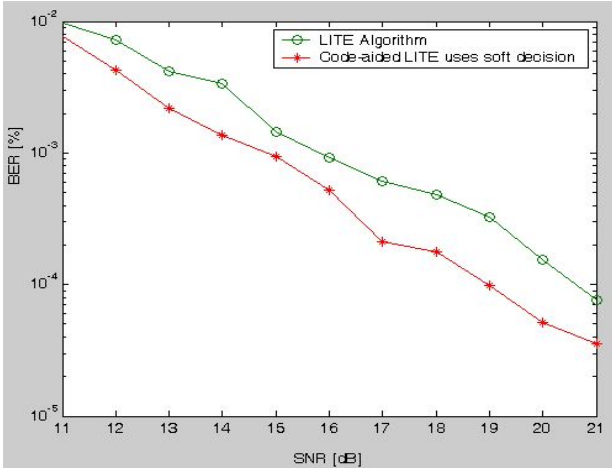

FIG. 5 THE BER PERFORMANCE UNDER A RAYLEIGH CHANNEL

Fig. 4 shows the performance analysis in the Additive

White Gaussian Noise (AWGN) channel and Fig. 5 shows the one in Rayleigh channel, respectively. In Fig. 5, the system performance using LITE algorithm without a coding scheme shows the worse performance than the one with code-aided LITE because the coding scheme helps to recover the desired signal. Similarly, the performance shows the results in a fading channel. Specially, code-aided LITE provides a benefit to the system performance in the fading channel.

Conclusion

The performance analysis using a code-aided linear equalizer in parallel fading channels is provided in this paper. Comparing with the proposed convolutional code-aided LITE and LITE without coding scheme, the system performance is improved in a fading channel, obviously. The proposed code-aided LITE is more suitable to be employed in a fading transmission channel to recover the transmitted data from the .distorted signals.

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